ABSTRACT

Audio data decoded in an MPEG system to be stored in a storage unit is supplied to an audio output via a filtering processing. For performing the filtering processing, presentation time interval of respective audio data is changed to conform to a user's designated playback speed, and the decoded audio data stored in the storage unit by being synchronized with the changed presentation time interval is written on an input queue in the set unit. A TSM algorithm is performed in the frame unit with respect to the audio data of the input queue to decrease the quantity of the audio data when the designated playback speed is faster than a normal playback speed or to increase it when the designated playback speed is slower than the normal playback speed, in accordance with a value of the designated playback speed. The TSM audio data is transferred to a middle queue. With respect to the audio data of the middle queue, up-sampling or down-sampling is performed in accordance with the value of the designated playback speed. The quantity of the audio data after the sampling becomes substantially the same as that of the decoded audio data, and thus the sampled audio data have a tone substantially identical to that of the normal playback speed and are transmitted to an output queue. The audio data stored in the output queue is synchronized with the changed presentation time interval to be transmitted to the storage in the set unit, and then is reproduced via an audio output.